



## **SSCA® Certification**

---

***Become a 'SIP School Certified Associate' endorsed by the Telecommunications Industry Association (TIA)***

### **Exam Objectives**

The SSCA® exam is designed to test your skills and knowledge on the protocol SIP (Session Initiation Protocol). Everything that you need to cover in order to pass this test is covered in the [SSCA® SIP training program](#) but if you decide to learn about SIP elsewhere then these are the topics that you should learn about in order to be prepared for the test.

This list is the same as the 'course topics' list also found under the 'outline' button next to the course name in the Catalog.

Please note that if you go along an alternate training path it is possible that you may get a question that may not have been covered in that path. It's up to you!

**Please view the following pages for the complete topic list....**

---

# Core SIP

---

Topics:

## **SIP – Who Benefits**

- SIP – The Session Initiation Protocol
- SIP 'Official Summary'
- Based on HTML
- Where does SIP fit in?
- SIP Clients and Servers
- SIP User Agents
- Simple Call Session Setup
- SIP System Architecture
- The URI - Unique Resource Identifier
- SIP Addressing
- SIP Addressing Examples

## **SIP Servers and Operation**

- Registration
- Re-Registration
- SIP Proxy servers and why we need them
- SIP Server – Proxy Mode
- SIP Server – Re-Direct Mode
- Proxy Server 'State' types
- Location Services
- Registration
- Re-Registration
- DHCP and SIP
- SIP Proxy – Trapezoid Model
- SIP Server in Proxy Mode
- SIP Server in Proxy Redirect Mode
- Stateful and Stateless Proxies
- Location Server
- Location Server – Components
- Location Server – Information Sources
- Location Server – Example

## **SIP Messaging**

- Request Methods
- Response Codes
- SIP Headers
- INVITE – Example
- RESPONSE – Example
- SIP Request Methods
- SIP Response Codes
- SIP Headers
- SIP HEADER - INVITE
- SIP HEADER - 200 Response

## **SDP – the Session Description Protocol**

- SDP – The Session Description Protocol
- SDP in a SIP Message
- An SDP Example
- Extending SDP
- Changing Session Parameters
- Call Hold example
- Multiple 'm' lines
- SDP – The Session Description Protocol
- SDP Component in a SIP Message
- SDP Example
- Extending SDP
- Changing Session Parameters
- SDP Example - Put a call on Hold
- SDP Example - Call Hold Trace

- Call Hold – Old and New Methods
- Music on Hold example
- INVITE and reINVITE

#### **SIP Mobility**

- SIP Mobility
- SIP Call Forking - Parallel
- SIP Call Forking - Sequential
- Call legs, dialogs and Call IDs
- Dialog trace example
- Dialogs and Transactions
- Branch Ids
- Call Forward - No Answer
- Call Forward to Voicemail

#### **More on Proxies and SIP Routing**

- Stateless Proxy
- Stateful Proxy
- More Proxy information
- VIA and Record Route
- VIA Details
- Record-Route Defined
- Record Route Example
- Loose and Strict Routing
- Session Policies

#### **MIME**

- MIME
- Multiple MIME parts

#### **SIP and the PSTN**

- SIP and the PSTN
- SIP to PSTN Call Flow
- SIP to PSTN Detail
- SIP Codes and the PSTN

#### **SIP and B2BUA**

- B2BUA - Back to Back User Agent
- B2BUA Example
- B2BUA Benefits and Features

#### **SIP Summary**

- Request for Comments
- New RFCs
- SIPIT
- The Call Process

# Wireshark

---

Topics:

## Wireshark

- What is Wireshark?
- Your Initial Setup
- Free @thesipschool.com account from OnSIP
- SIP account with Voipuser.org
- X-lite client for testing
- Configure X-Lite
- SIP Social directory
- SIP test numbers
- Download Wireshark
- Wireshark – Basic Layout
- Wireshark icons
- Using Wireshark – Capturing
- Using Wireshark – Simple Filters
- Using Wireshark – More SIP statistics
- Using Wireshark – RTP Statistics
- Saving Captures
- What are the codes?
- Link to Troubleshooting module for Advanced Wireshark

# SIP, the PSTN and SIP-T

---

Topics:

## **SIP-T and the PSTN**

- SIP to PSTN Overview
- SIP to PSTN Call Flow
- SIP to PSTN Detail
- PSTN to SIP Call Flow
- SIP to PSTN Call Failure
- SIP to PSTN Call trace

## **Early Media**

- Early Media - SIP to PSTN Call

## **Early Offer and Delayed Offer**

- Early Offer / Delayed Offer

## **Gateways**

- Default Gateway?
- Gateway Location and Routing with TRIP
- TRIP Examples

## **SIP-T and PSTN Bridging**

- SIP-T
- SS7, ISDN and SIP
- ISUP and SIP Messages
- ISDN User Part (ISUP) to SIP Codes
- PSTN to PSTN via SIP
- ISUP Encapsulation
- ISUP Encapsulation / SDP
- Addressing Notes

## **SIP and DTMF**

- DTMF - Quick Re-Cap
- What is DTMF?
- DTMF Transport methods
- DTMF 'Inband'
- RFC 2833 'Trace' example
- RFC 4733 replaces 2833
- RFC 4734
- SIP INFO 'Trace' example

# SIP, VoIP and QoS

---

Topics:

## **What is VoIP or Voice over IP?**

- What is VoIP?
- What is Voice over IP?
- VoIP – 'A Basic Call'
- VoIP and TCP / UDP
- VoIP over the Internet
- Branch to Branch VoIP
- IP PBX

## **Voice Sampling and Codec**

- Encoding
- Codecs for Voice
- Try the Codec Test
- High Definition (HD) Voice
- Sound tests
- Wideband (HD) codecs
- MOS – Mean Opinion scores

## **The Real time Protocol or RTP**

- RTP Encapsulation
- RTP Header Trace
- Real Time Control Protocol
- RTCP-XR (Extended Reports)
- RTP / RTCP and UDP Ports

## **Quality of Service**

- QoS basics
- QoS Issues
- Measuring Delay
- Jitter and Packet Loss
- General VoIP Acceptance Criteria
- QoS on the Network
- 802.1Q – VLANs
- 802.1Q/P Tagging
- 802.1P - L2 Classification
- TOS and DiffServe
- Layer 3 Classification
- Codecs and Bandwidth
- Symmetric DSL (SDSL)
- Testing your link

## **SIP, SDP and VoIP**

- SIP in the TCP/IP Model
- SIP and SDP Messages
- SIP and SDP Codec mapping
- Where does SIP fit in?
- SIP, SDP and VoIP INVITE
- Audio and Video in the SDP body

# SIP Security

---

Topics:

## Authentication and Authorization

- SIP Proxy Authentication
- 401 and 407 Authorization
- SIP Authorization
- PROXY Authentication
- SSL with MD5 Cracked!
- MD5 v SHA

## Encryption

- Why Encrypt SIP?
- Certificates and HTTPS
- Certificate Authorities
- Certificate Example
- Self-Signed Certificates
- Format type
- Securing SIP and VoIP
- SSL and TLS
- SIP and TLS
- TLS Thoughts
- TLS and SIP in Action
- SIPS and SIP Addressing
- Secure RTP (SRTP)
- Setting SRTP on SIP Devices
- Secure RTP (SRTP) - Example
- SRTP and SRTCP
- sdes and the Crypto attribute
- Crypto attribute example
- Crypto multiple streams
- RFC 4474 for Caller Identity
- Caller Identity
- DTLS/SRTP
- S/MIME and SIP
- MIME and ISUP
- SIP Trunking and Security
- Enhancing SIP Trunk Security
- Alternatives - IPSec, ZRTP

## Attacks and Responses

- Types of Attack on a VoIP/SIP Network
- FBI network examples
- Responses and Protection
- TLS v SSL
- Response Identity – A Problem!
- Rogue SIP Proxy
- Phishing and SIP exploit
- More Examples RFC 4475
- Try for yourself with recommended software tools

## NIST Recommendations

- NIST Recommendations on securing VoIP

# Firewalls, NAT and Session Border Controllers

---

Topics:

## Overview

- Issues to address

## Firewalls

- What does a Firewall do?
- Are Firewalls effective?

## NAT or Network Address Translation

- What is NAT?
- NAT Request
- NAT Response
- Multiple NATs
- The NAT Problem

## Types of NAT

- Types of NAT
- NAT – Full Cone
- NAT – Restricted Cone
- NAT – Port Restricted Cone
- NAT – Symmetric
- The NAT or (PAT) Problem
- Problems with NAT, Firewalls and SIP

## The Solutions

- STUN (Simple Traversal of UDP)
- STUN (Simple Traversal of UDP)
- STUN and rport
- Problems with STUN
- TURN (Traversal Using Relay NAT)
- Interactive Connectivity Establishment (ICE)
- How ICE works – Simplified!
- More on ICE
- Universal Plug and Play (UPnP)
- Near end NAT
- Far end NAT

## The RTP Problem

- The Firewall Problem
- Solving the RTP Problem
- Symmetric RTP
- Media Proxy
- Application Level Gateway
- SIP Aware Firewalls - Incoming
- SIP Aware Firewalls - Outgoing

## Session Border Controllers

- SBC for the Enterprise
- SBC for the ITSP
- Recommended Session Border Controller features
- SBCs in Action!

# SIP Trunking

---

Topics:

## **SIP Trunks**

- What is a SIP Trunk
- Alternative to TDM
- Separate Data and Voice connections
- Converging the network
- SIP Trunks and Codec
- SIP Trunk Benefits

## **SIP Trunking – In More Depth**

- SIP Trunk Capabilities
- SIP Trunking Network Examples
- SIP Peering
- Peering problems?
- Least Cost routing (LCR)
- Disaster Recovery
- Disaster Recovery 'Expanded detail'
- Disaster Recovery – Last resort?
- Number Consolidation
- Virtual Presences

## **Trunking Variations**

- Single Site, TDM PBX
- Single Site, No 'Forklift'
- Single Site, Converged
- Converged – SIP/IP PBX
- Multiple Site, 'Converged'

## **Media Gateways**

- SIP PBX to Non-SIP PBX
- SIP PBX to Non-SIP PBX, Call Flow

## **SIP Trunk Performance**

- The ADSL issue
- Codecs, Voice and Data
- Symmetric DSL (SDSL)
- Bandwidth Calculator
- Testing your link
- ADSL Developments
- Fibre Options

## **SIP Trunking and MPLS**

- MPLS, basic explanation
- MPLS Label format
- MPLS in a MAC frame
- MPLS example network
- MPLS benefits
- Your own private WAN
- but 'Not the only client'
- Separate MPLS networks

## **Security and SIP Trunking**

- SIP Trunk Security - Overview
- Session Border Controllers

## **More on SBCs**

- The 'corporate' SBC
- SIP REFER issues

## **Setting up a SIP Trunk**

- Add a VoIP Provider

- Provider SIP Servers
- Authentication
- Add a Dialling Rule
- Trunk setup complete
- Call out Trace
- Comparing SIP packets from two ITSP providers

#### **Some PBX Requirements**

- Enterprise PSTN Identities
- P-Preferred and P-Asserted
- Call Progress Tones

#### **Next Generation Networks**

- What are NGNs?
- An Example – British Telecom

#### **Troubleshooting and Interops**

- SIP Trunks and Common Problems
- The SIP Forum
- SIPits
- SIPit Results
- SIP Connect Document
- SIP Connect 1.1

#### **Choosing an ITSP**

- Understanding ITSP Offerings
- 'Sticking points'?

#### **Resource Websites**

- TMCnet – Sip trunking
- Siptrunk.org
- No Jitter - Hotzone

# Testing, Troubleshooting and Interoperability

---

## Setting up your test environment

- Using SIP IP Phones
- Using SIP Softphones
- Even more SIP Softphones
- SIP Communicator
- Choosing a 'Trial/Test' ITSP
- Getting Free ITSP Accounts
- Configuring your Softphone
- Get a SIP URI of your own
- SIP2SIP accounts
- Configuring SIP Communicator with a SIP2SIP account
- Using 'Test Numbers'
- Multiple Setup options for you to try
- Configure X-Lite and SIP Communicator on the same PC for testing
- Example - The SIP Phones @ The SIP School™

## Wireshark

- Loading Wireshark
- Network interface setup for capture
- Wireshark - Basic Layout
- Understanding Wireshark Icons
- Using Wireshark - Capturing
- Using Wireshark – Simple Filters
- Using Wireshark – SIP Statistics
- Saving Captures
- Wireshark in more depth!
- SIP Statistics
- RTP / VoIP Capture and Playback
- More 'SIP ladder' analysis
- Coloring rules
- More 'filter expressions'
- More Help on Wireshark if you need it
- You try
- Where to Capture?

## Interoperability Testing

- Interop Testing
- Why Interop can be tough
- Different interpretations in the RFC 3261
- BLISS – Basic Interoperability for SIP Services
- Interop Test Scenario
- Interop Test operations
- Sample Interop Traces
- Wireshark example videos to help understand interop issues
- Video call testing
- Video tests with Wireshark trace analysis
- SIPIT events

## Common SIP problems

- Will it ever work?
- What else can you do?
- Common SIP/VoIP Problems
- Troubleshooting SIP Trunks
- 4xx — Client Failure Responses
- 5xx — Server Failure Responses
- 6xx — Global Failure Responses

## More SIP Testing Tools

- SIP Scenario
- SIP Workbench
- SIP Monitoring example app
- SIP Scan

- [TestYourVoIP.com](http://TestYourVoIP.com)
- [HoverIP](http://HoverIP)
- [NSLookup](http://NSLookup)
- [SIP Center and Voip-info for more tools!](#)
- [Using the NET to find answers](#)
- [The SIP Wiki](#)

# ENUM, DNS and VoIP Peering

---

Topics:

## ENUM Explained

- What is E.164?
- What is ENUM?
- Why ENUM?
- Call Routing and ENUM - Example

## Enum, DNS and Domains

- Why are we using DNS?
- DNS and the Web
- 'Finding' Domain name servers using NSLookup
- The e164.arpa Domain
- Approved ENUM Delegations
- TIERS 0, 1, 2 and 3
- TIERS and Registrars
- DNS and AOR
- e164.arpa Domain in action
- Example - ENUM in the UK
- Address of Record
- Reseaux IP Europeens
- PSTN to SIP UA - Example
- The ENUM Query
- NAPTR Records
- DNS Response to an ENUM query
- 'Finding SIP servers using the tool - DIG
- Calls Flows
- PSTN to SIP UA – Example (2)
- IP to PSTN (Simplified)
- MARTINI

## Types of ENUM

- Different 'Types' of ENUM
- The Problems with 'Public' ENUM
- Example – 'Private' ENUM
- Example – 'Operator' ENUM

## VoIP Peering

- Stay 'On-Net'
- From ITSP to PSTN and Back...!
- Loss of features with the PSTN
- Peering Profiles and Agreements
- Bi-lateral Peering
- Multi-lateral Peering
- A few providers
- ENUM in North America
- A complete 'infrastructure'

## ViPR

- Verification Involving PSTN Reachability (ViPR).
- What is ViPR
- ViPR and P2P
- Initial PSTN Call
- ViPR Call Record
- Query the DHT
- DHT query and Validation
- The Next call is a SIP Call
- ViPR Summary

## Try for yourself

- Register your number

- Testing ENUM
- DIG and NAPTR records
- ENUMPlus for multiple ENUM trees

**ENUM and the future**

- How is ENUM moving forward?
- Useful Links

# SIP and Fax over IP

---

Topics:

## **Faxing Basics**

- Faxing background
- T.30 Fax signaling
- Associated tones and protocols
- The ITU and TIA standards

## **Fax over IP**

- Fax over IP benefits
- From the old to the new
- Intro to FoIP
- FoIP and SIP trunks
- Protocol conversions

## **Fax Protocols**

- G.711 Pass-through
- T.37 Store and Forward
- T.38 Relay
- Where does SIP fit in?
- UDPTL

## **FoIP in action**

- SIP in FoIP – Call Flow
- SIP INVITE
- INVITE for T.38
- The INVITE SDP body
- Wireshark FoIP example
- SIP T.38 Call flows – IETF draft document

## **Bandwidth**

- T.38 and G.711 network traffic

## **Troubleshooting**

- The basics
- More complex issues to watch out for

## **Ongoing Efforts**

- SIP Forum and i3 Forum
- Use DTMF events instead?

# SIP and Unified Communications

---

## Topics Include

### Communication Breakdown

- Playing Voicemail tag
- Can't find people
- Available but not Available..!
- More Examples of communication problems

### IM Clients

- IM Client Features
- Enterprise Clients
- More in IM Clients
- IM and Mobile devices

### The Background Stuff

- The IMPP working group
- IMPP and CPP
- More IMPP work
- SIMPLE

### How it all works

- Presentity
- A Basic SIP subscription
- Multiple Presence States
- Presence and P2P
- A Presence Network
- Getting inside the SIP packets
- Presentity and more!
- A Basic SIP Subscription
- Multiple Presence States
- Presence and P2P
- A Presence Network
- Get inside the SIP packets
- The Packet Structure
- PIDF Message Body
- XML
- Tuples
- Example Presence doc with Tuples (using a Mobile Phone)
- Rich Presence
- The METHODS in Action
- PUBLISH STATE
- PUBLISH and PIDF/XML body
- SUBSCRIBE METHOD
- 202 OK Response
- NOTIFY
- MESSAGE
- Add A Buddy/Subscribe
- is-composing
- Alternative 'Presence States'
- 2 Places at the same time

### Conferencing

- What SIP does in Conferencing
- INITIATE a conference
- JOIN a conference
- LEAVE / EXIT a conference
- INVITE other participants
- REFER conference server to invite or others to join
- EXPEL participants
- CONFIGURE the media stream
- CONTROL a conference

- Why SIP?
- Centralized conferencing
- Centralized Signaling
- Centralized Mixing (optional)
- Centralized Authentication
- B2BUA (Discussed in core module)
- Conference Components
- The Focus
- More than one Focus
- Conference Setup
- iscomposing in Conference
- MESSAGE in conference
- BYE in conference
- Alternative INVITE
- SDP BODY OF INVITE
- IETF work and Conferencing

#### **XMPP v SIMPLE**

- What is XMPP?
- SIMPLE and/or XMPP
- Gateways

#### **Federations**

- What is Federation?
- Multiple Presence sources
- Super-Aggregation
- Inter-Domain Federation

#### **Unified Communications**

- What's all the fuss?
- Unified Confusion
- Components involved
- What should UC do?
- 21<sup>st</sup> Century Dial tone
- The Unified inbox
- Unified aware applications
- Find me – Follow me
- Device awareness
- Unified Comms for Business
- Do your Homework
- Humans and UC
- UC in a SIP network

#### **UCI Forum**

- The UCI Forum - Challenges
- UCI Forum goals
- UCIF website

#### **Relevant RFCs**

- RFCs Galore